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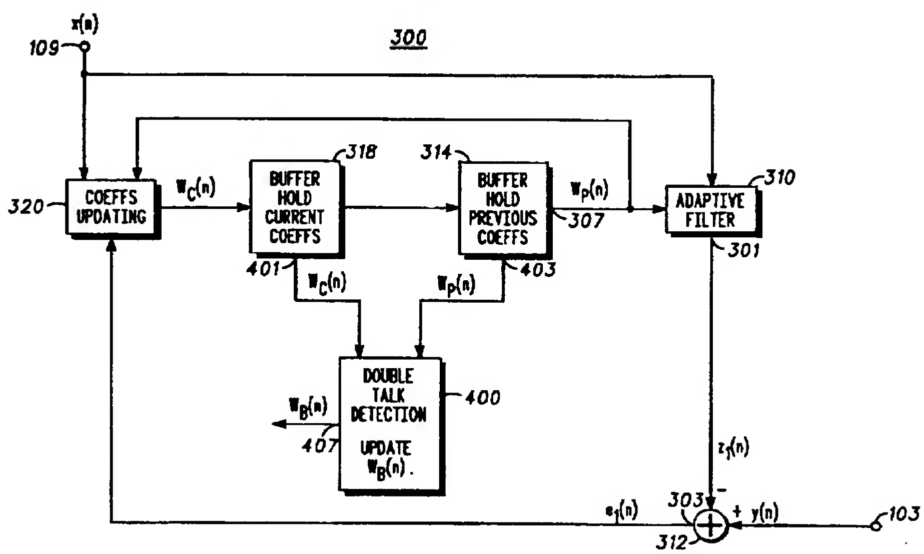
(54) Abstract Title

Detecting double-talk in an echo canceller

(57) A double-talk detection device (400) detects the presence of near-end speech signals for an echo canceller including an adaptive filter to generate an echo estimate. A control device (300) generates the coefficients for the adaptive filter. The control device (300) includes a coefficient updating circuit (320). A buffer (314) holds the previous filter coefficients and a buffer (318) holds the current filter coefficients. The double-talk detection device (400) detects the presence of near-end speech signals based upon a detected variation of the coefficients. The detection device (400) is used to detect a double-talk condition during a single-talk condition and a single-talk condition during the double-talk condition.



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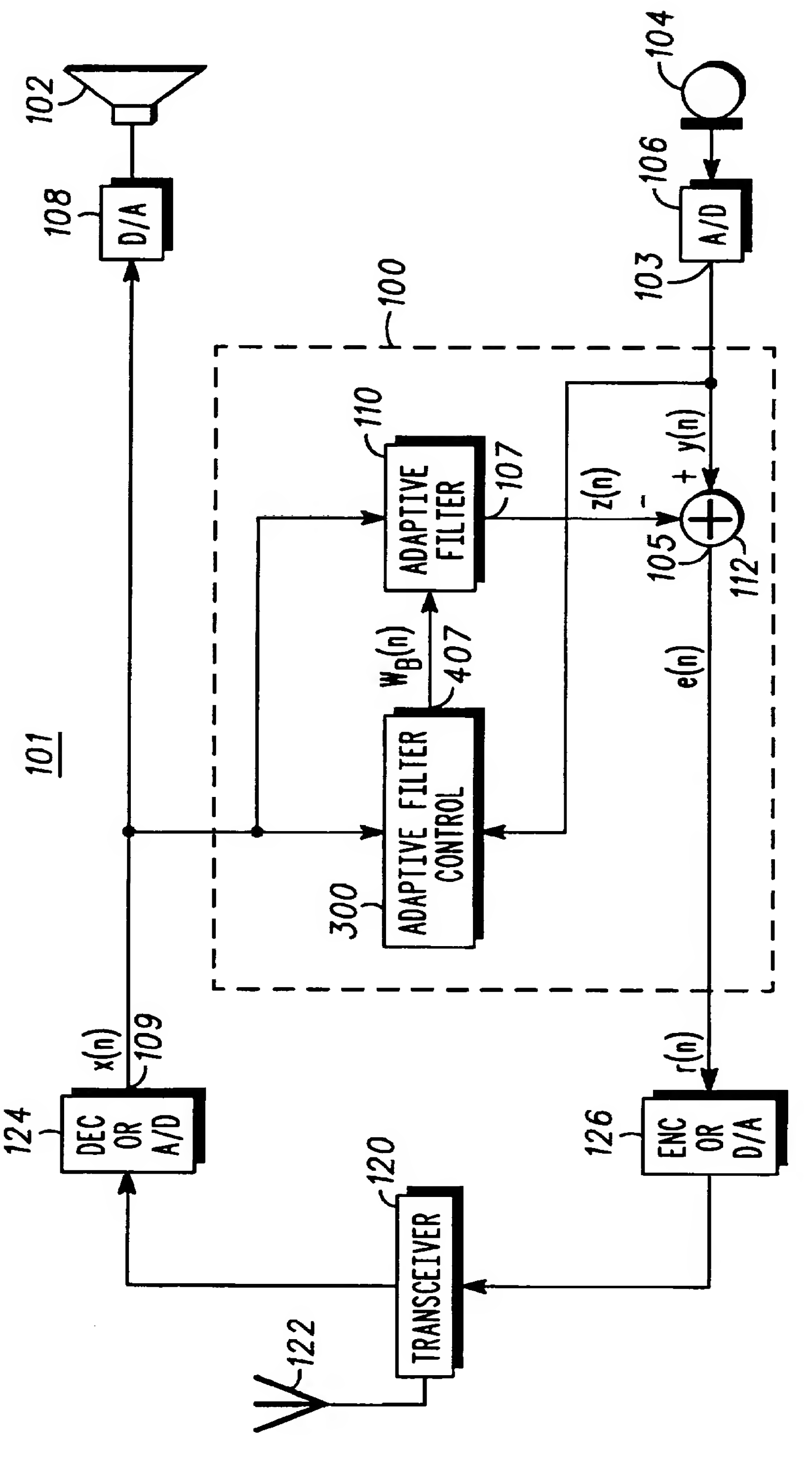
101
102
108
D/A
104
106
A/D
103
100
110
ADAPTIVE FILTER
107
z(n)
y(n)
105
112
$W_B(n)$
407
e(n)
300
ADAPTIVE FILTER CONTROL
x(n)
109
124
DEC OR A/D
r(n)
126
ENC OR D/A
120
TRANSCEIVER
122

FIG.1

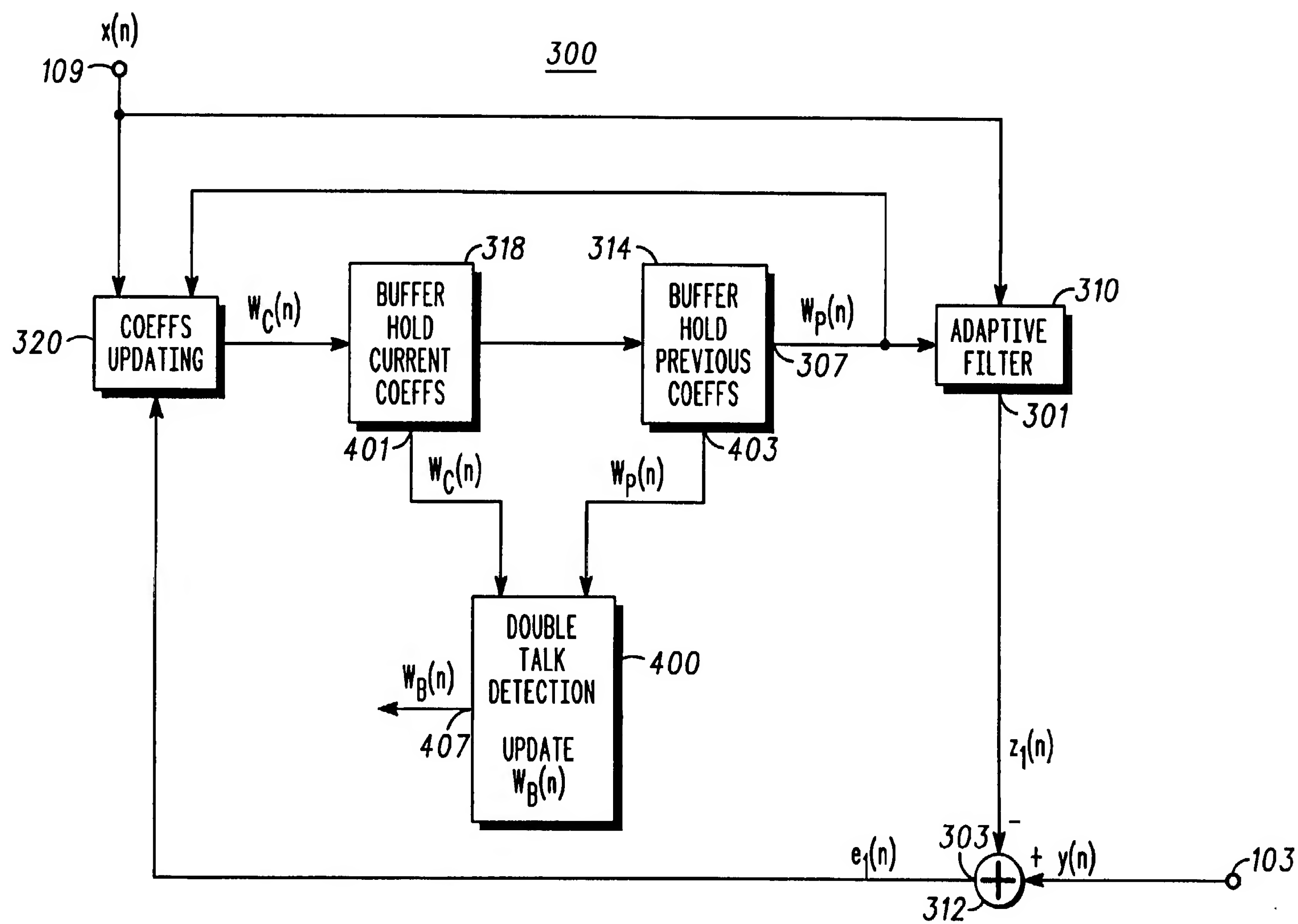
x(n)
109
300
320
COEFFS UPDATING
$W_C(n)$
318
BUFFER HOLD CURRENT COEFFS
314
BUFFER HOLD PREVIOUS COEFFS
$W_P(n)$
307
310
ADAPTIVE FILTER
301
401
$W_C(n)$
$W_P(n)$
403
DOUBLE TALK DETECTION
UPDATE $W_B(n)$
400
$W_B(n)$
407
$z_1(n)$
303
−
+
y(n)
103
$e_1(n)$
312

FIG. 2

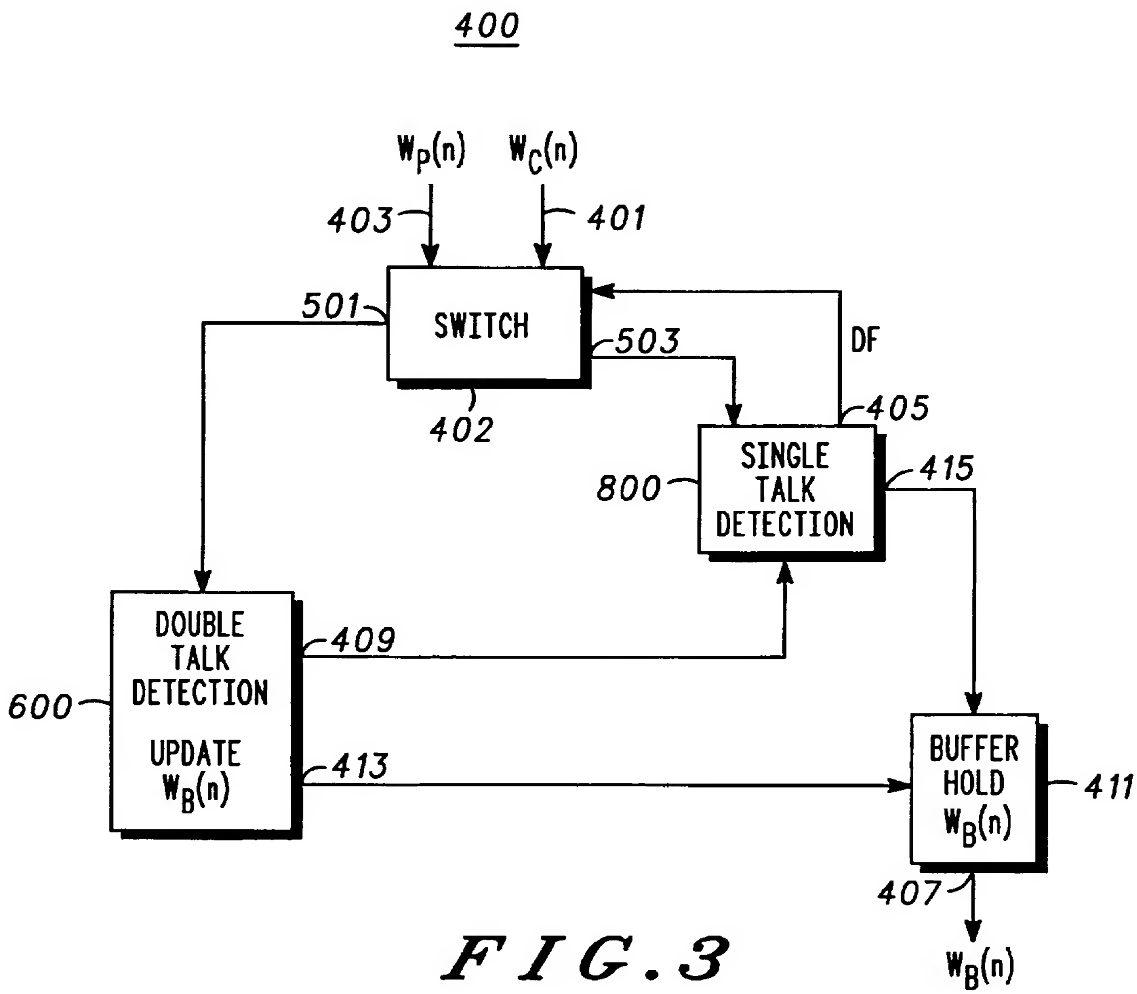
400
$W_P(n)$
$W_C(n)$
403
401
501
SWITCH
503
DF
402
405
SINGLE TALK DETECTION
800
415
DOUBLE TALK DETECTION
UPDATE $W_B(n)$
409
600
413
BUFFER HOLD $W_B(n)$
411
407
$W_B(n)$

FIG. 3

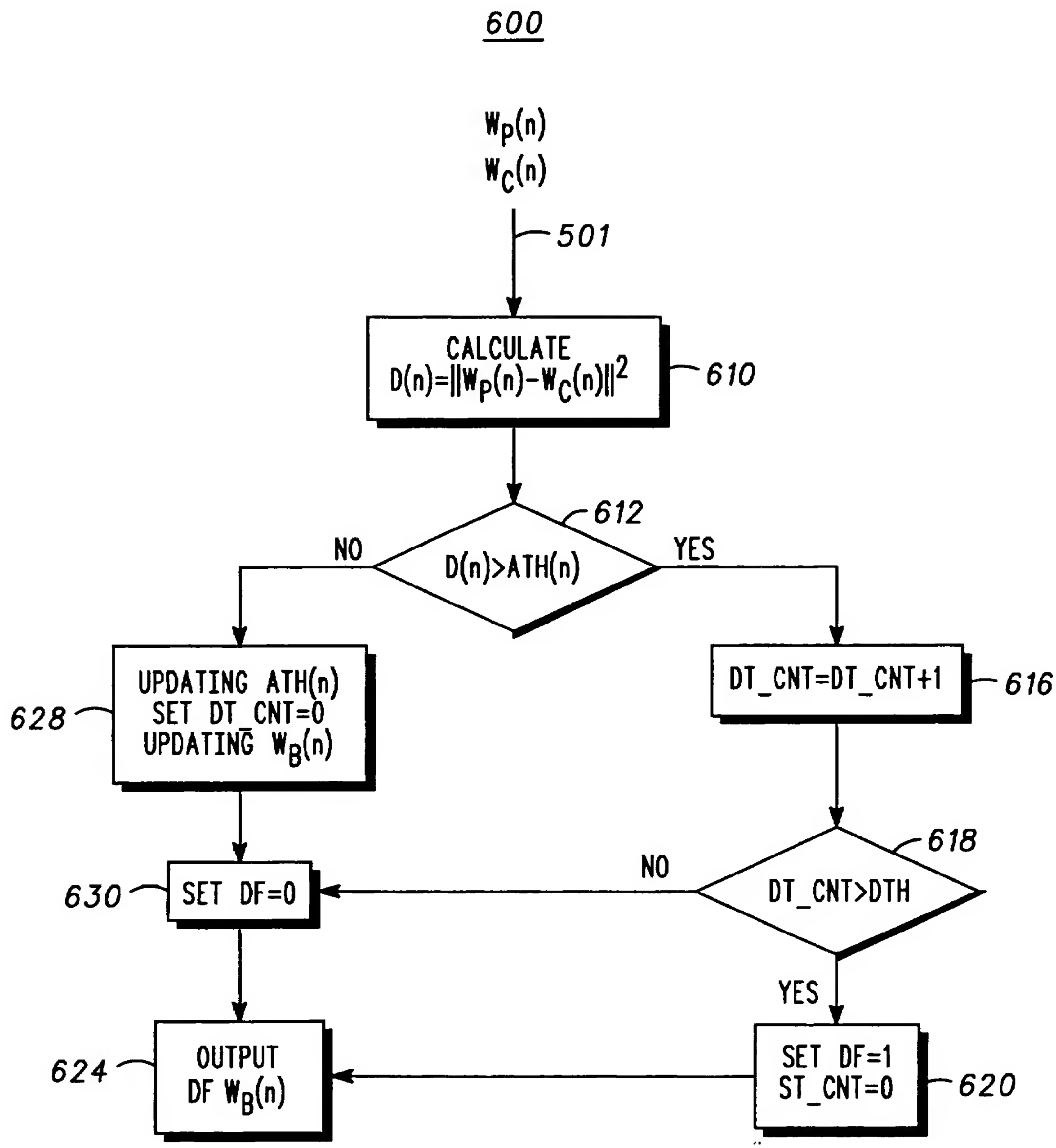
600
$W_P(n)$
$W_C(n)$
501
CALCULATE
$D(n)=\|W_P(n)-W_C(n)\|^2$
610
612
NO
D(n)>ATH(n)
YES
UPDATING ATH(n)
SET DT_CNT=0
UPDATING $W_B(n)$
628
DT_CNT=DT_CNT+1
616
618
NO
DT_CNT>DTH
630
SET DF=0
YES
OUTPUT
DF $W_B(n)$
624
SET DF=1
ST_CNT=0
620

FIG.4

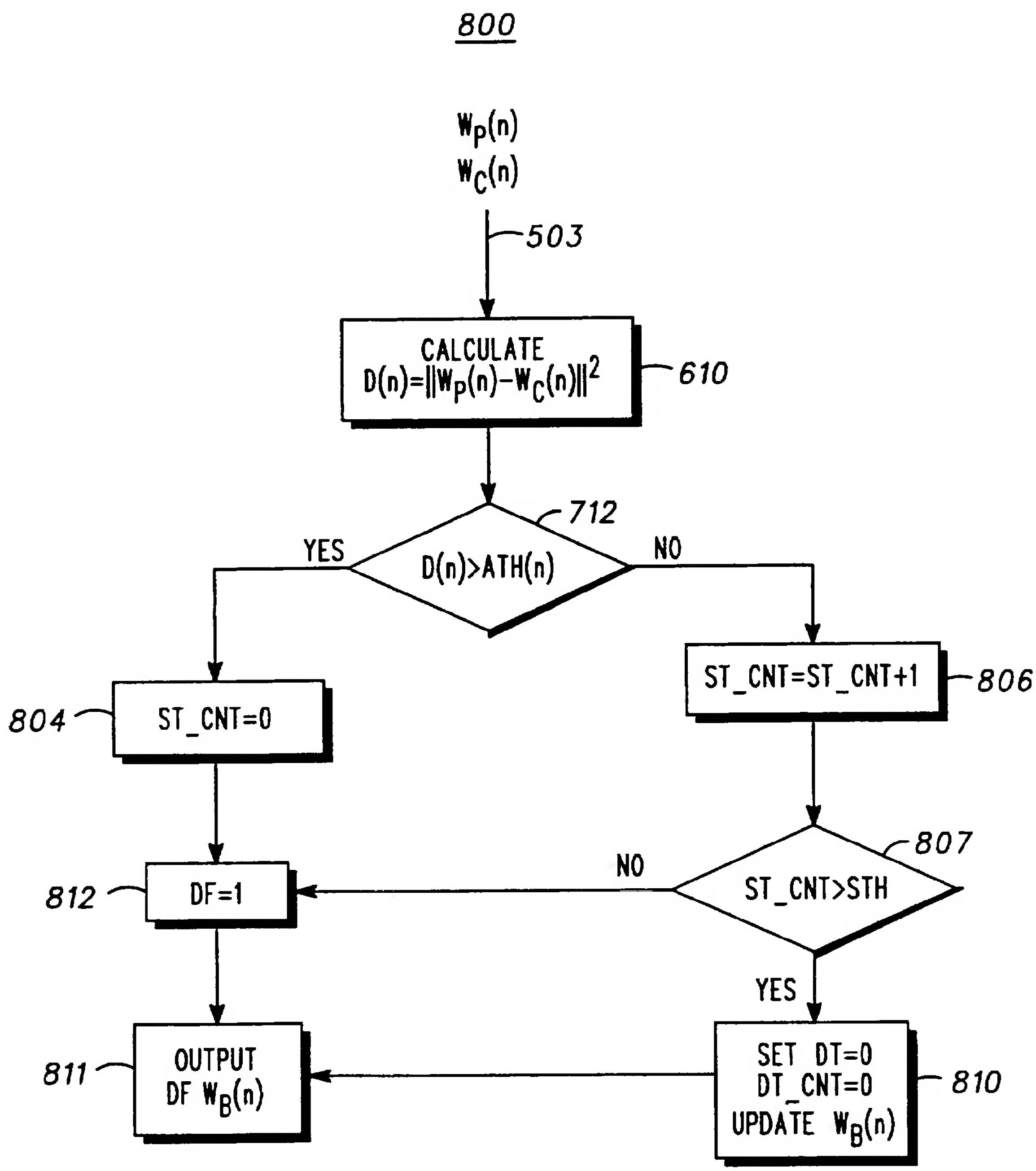
800
$W_P(n)$
$W_C(n)$
503
CALCULATE
$D(n)=\|W_P(n)-W_C(n)\|^2$
610
712
YES
$D(n)>ATH(n)$
NO
ST_CNT=ST_CNT+1
806
804
ST_CNT=0
807
NO
812
DF=1
ST_CNT>STH
YES
811
OUTPUT
DF $W_B(n)$
SET DT=0
DT_CNT=0
UPDATE $W_B(n)$
810

FIG.5

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METHOD AND DEVICE FOR DETECTING NEAR-END VOICE

FIELD OF THE INVENTION

The present invention pertains generally to communication systems and more particularly to double-talk detection for an adaptive echo cancellation system in a communication system.

BACKGROUND OF THE INVENTION

The need for echo cancellation arises in many full-duplex communication systems. One particularly challenging environment where the need for reliable echo cancellation exists is full-duplex hands-free operation of cellular radiotelephone devices and teleconferencing devices. During hands-free operation of such devices, signals from the speaker are fed back into the microphone through various acoustic paths and are subject to delay before reaching the speaker such that they are perceived by far-end users as echo signals. These echo signals, commonly referred to as acoustic echo, are very annoying to the participants involved in two-way communication and difficult to eliminate.

One of the most effective solutions generally used for eliminating echo signals employs echo cancellers having an adaptive filter. The least means square (LMS) adaptive filter is the most common type of filter used. An LMS filter is a finite impulse response (FIR) filter which models an echo path through adaptively adjusted coefficients. The coefficients of the filter are adaptively trained using the far-end signal, which drives the loudspeaker of a hands-free communication device, and the near-end signal, which is output from the microphone of the hands-free communication device. In a hands-free device, the adaptive filter adaptively synthesizes a replica of the echo from the far-end signal, which replica is subtracted from the received signal output by the microphone at the near-end. The result is a substantially echo-free signal which is further transmitted to the far-end.

There are four operating states, or conditions, for a bi-directional communication device, which states will be described herein with reference to a hands-free communication device: a near-end single-talk state which occurs when only near-end speech is present; a far-end single-talk state which occurs when only far-end speech is present; a double-talk state which occurs when near-end and far-end speech signals are both present; and a no-talk sate which occurs when neither near-end nor far-end speech is

present. It is well understood that adaptation of the filter coefficients is best performed during the far-end single-talk state.

When a double-talk condition occurs, adaptation must stop. If adaptation does not stop, the filter coefficients will diverge from their optimum value. On the other hand, a high false detection rate will disable coefficient adaptation frequently during the single-talk condition, and thus significantly slow down the convergence speed of the adaptive filter coefficients. This is a particularly serious problem in applications involving a fast-changing echo path and high noise environments, such as hands-free radiotelephones and teleconferencing environments. Therefore, a reliable double-talk detector is an important component for successful echo cancellation.

Several double-talk detectors have been implemented that are based upon signal energy measurements or measured signal correlation. The signal energy based methods detect the double-talk condition based upon the relationship of the signal energy in the transmit and receive paths, which differs during the double-talk state and the single-talk state. For example, one of the simplest implementations compares the energy of the far-end signal to the energy of the near-end signal. In general the near-end signal energy increases in the transition from a single-talk condition to a double-talk condition. The echo residual signal energy also increases during the transition from the single-talk condition to the double-talk condition. By examining the far-end signal and either the near-end signal energy or the echo residual signal energy, one can detect a double-talk condition.

The echo-return-loss ratio has also been used to detect a double-talk condition. The echo-return-loss ratio is commonly defined as the ratio of the echo residual energy to the near-end echo signal energy. The echo-return-loss ratio increases during the transition from a single-talk condition to a double-talk condition. Other methods combine various signal energies for double-talk detection. The problems experienced with the energy-based methods include that they are not sensitive enough to low-level near-end speech signals and they produce a high number of false detections if the near-end is a noisy environment (near-end being the hands-free environment where the device is located and the far-end being the location of a participant connected through a communication channel such as a telephone line).

Correlation-based methods perform double-talk detection based upon the signal correlation of the far-end and near-end signals. The correlation of

the far-end signal and the near-end signal is normally higher in a single-talk condition than a double-talk condition. By comparing the correlation level of the far-end signal and the near-end signal to a threshold, the condition can be labeled a double-talk condition or single-talk condition. Another implementation uses the correlation of the far-end signal and the echo residual signal. In such systems, a double-talk condition is identified when the echo residual signal is not well correlated to the far-end signal. In either case, the double-talk detection is based upon an assumption that correlation of signals in the transmit and receive paths of a bi-directional communication device is lowered when there is a transition from a single-talk condition to a double-talk condition.

It is also known to monitor the correlation of the echo estimate signal and the echo residual signal to detect a double-talk condition. Normally, this correlation increases in the transition from a double-talk condition to a single-talk condition. However, because correlation is sensitive to noise, these techniques can not be used in noisy environments.

Accordingly, a need remains for a more robust and precise double-talk detector that will operate in all environments, including environments subject to high amplitude and large dynamic range background noise and rapidly changing echo paths.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a circuit schematic illustrating a full-duplex communication device employing an adaptive echo canceller in which the present invention is used.

FIG. 2 is a circuit schematic illustrating a control device for an adaptive echo canceller.

FIG. 3 is a circuit schematic illustrating a double-talk detection device.

FIG. 4 is a flow chart illustrating operation of double-talk detector in single-talk condition.

FIG. 5 is a flow chart illustrating operation of single-talk detector in double-talk condition.

DETAILED DESCRIPTION OF THE DRAWINGS

A double-talk detector is associated with an echo canceller having an adaptive filter with adjustable coefficients. The echo canceller includes an adaptive filter to generate an echo estimate, a subtractor to generate an echo-cancelled signal for transmission, and a control device to update the coefficients for the adaptive filter and to perform double-talk detection. The control device includes a buffer storing current coefficients for the adaptive filter, a buffer storing previous coefficients for the adaptive filter, a filter to generate an echo estimate using the previous filter coefficients, a subtractor to generate an echo estimate error signal used for updating filter coefficients, a filter coefficient updating device, and a double-talk detection device which selectively updates the filter coefficients for echo cancellation.

The double-talk detection device includes a double-talk detector and a single-talk detector. The double-talk detector monitors the coefficient variation, and may for example detect the difference between the current coefficients and previous coefficients. If the difference is too large over a predetermined period, a double-talk condition is detected. An adaptive variance threshold can be employed. The single-talk detector detects when the difference between current coefficients and previous coefficients drops below a variance threshold. If the difference drops below the variance threshold for a predetermined period, a single-talk condition is detected. An adaptive threshold can be employed, and the thresholds for both single-talk and double-talk detection can be generated in the same manner.

FIG. 1 shows a full-duplex communication device 101 employing an echo canceller 100 in which the present invention can be used. The full-

duplex communication device 101 is illustrated as a hands-free device, and may be a hands-free radiotelephone, a hands-free teleconferencing device, a hands-free satellite telephone, a hands-free cordless telephone, a personal computer (PC) multimedia communication device, or any other suitable communication device. Additionally, those skilled in the art will recognize that a double-talk detection device as disclosed herein can be used in other environments having echo signals, such as repeaters, two-wire to four-wire converters, and the like.

The transmit path of the full-duplex communication device 101 includes a microphone 104 to pick up local (near-end) voice signals, an analog-to-digital (A/D) converter 106 to convert near-end voice signals from analog format to digital format, a subtractor 112 to subtract an echo estimate from the digitized near-end voice signal, and a digital-to-analog (D/A) converter 126 to convert echo-free transmit signals from the digital format to the analog format.

The receive path of the full-duplex communication device 101 includes an A/D converter 124 to convert far-end voice signals from an analog format to a digital format, a D/A converter 108 to convert far-end voice signals from digital format to an analog format, and a loudspeaker 102 to deliver remote (far-end) voice signals to local listeners.

A transceiver 120 is coupled to A/D converter 124 and D/A converter 126. The transceiver can be any suitable transceiver for cable, optical, wire-less, wire line or satellite communication, the operation of which are well known to those skilled in that art and are not described in greater detail herein for brevity. In the illustrated embodiment, transceiver 120 is coupled to antenna 122 for wireless communication in a cellular system. Transceiver 120 transmits near-end signals output by D/A converter 126 to a far-end communication device via antenna 122 and inputs received signals detected by antenna 122 to A/D converter 124.

Those skilled in the art will recognize that the A/D converter 124 and D/A converter 126 are used in an analog system. Alternatively the A/D converter 124 may be replaced by a speech encoder, and the D/A converter 126 may be replaced by a speech decoder, in some applications. For example, converters 124 and 126 can be a speech decoder and encoder of a digital interface in a communication device for the global system for mobile communications (GSM) or an integrated services digital network (ISDN). It is

further noted that the combination of antenna 122 and transceiver 120 may be replaced by a network interface device in some applications, such as applications for wire line or optical communication systems.

The far-end voice signal, x(n), in digital format at output 109 of the A/D converter 124, is input to the D/A converter 108, which generates an analog signal to drive the loudspeaker 102. A portion of the far-end signal output by speaker 102 is detected by microphone 104 with the near-end voice signal and converted to digital format in the A/D converter 106. The resulting near-end signal y(n) at an output 103 is input to the subtractor 112. The echo estimate signal, z(n), in digital format, is output from the adaptive filter 110 at output 107, and input to the subtractor 112. The echo-cancelled speech signal, e(n) or r(n) in digital format, is output at output 105 of the subtractor 112 and input to the D/A converter 126.

The adaptive echo canceller 100 can be implemented in a digital signal processor (DSP), a microprocessor, a programmable logic device, or the like. The echo canceller 100 includes an adaptive filter 110, which is illustrated to be a finite impulse response (FIR) filter with adjustable coefficients $\mathbf{W}_B(n)$ that are used to generate an echo estimate z(n). At the current sampling instant, n, a far-end speech sample x(n) is received from output 109 as the output of an A/D converter 124, and a near-end speech sample y(n) is received from output 103 as the output of A/D converter 106. Signals x(n) and y(n) are synchronized as A/D converter 106 and D/A converter 108 use the same clock. The echo estimate z(n) can be calculated based on the following equation:

$$z(n) = \mathbf{W}_B(n)^T \mathbf{X}(n) = \sum_{i=0}^{L-1} w^B_i(n)\, x(n-i) \qquad (1)$$

where superscript T means the transpose of a vector or a matrix, L is the order of the FIR adaptive filter 110 (and thus the number of coefficient positions in the adaptive filter 110), $\mathbf{X}(n) = [x(n)\ x(n-1)\ \ldots\ x(n-L+1)]^T$ holds L most recent far-end speech samples, $\mathbf{W}_B(n) = [w^B_0(n)\ W^B_1(n)\ \ldots\ w^B_{L-1}(n)]^T$ are the filter coefficients for echo cancellation. $\mathbf{W}_B(n)^T \mathbf{X}(n)$ is defined as the dot product of two vectors $\mathbf{W}_B(n)$ and $\mathbf{X}(n)$ as follows:

$$\mathbf{W}_B(n)^T \mathbf{X}(n) = \sum_{i=0}^{L-1} w^B_i(n)\, x(n-i).$$

The adaptive echo canceller 100 further includes a subtractor 112 which generates an echo-cancelled signal e(n) at output 105, for further transmission, by subtracting the echo estimate z(n) at output 107 from the near-end signal y(n) at output 103, such that:

$$e(n) = y(n) - z(n). \qquad (2)$$

The adaptive echo canceller 100 further includes a control device 300 to output the coefficients $\mathbf{W}_B(n)$ at output 407 based upon the far-end speech signal x(n) from an output 109 and the near-end speech signal y(n) from an output 103, and to perform double-talk detection.

Referring to FIG. 2, the control device 300 is coupled to output 109 to receive a far-end signal x(n), output 103 to receive a near-end signal y(n), and includes output 407 to provide the updated filter coefficients $\mathbf{W}_B(n)$. The control device 300 includes a coefficient updating circuit 320. A first buffer 318 stores the current coefficients, the current coefficients being generated during the most recent sampling interval. A second buffer 314 holds the previous coefficients, which are the coefficients for the sampling interval just prior to the most recent sampling interval.

A double-talk detection device 400 outputs the coefficients $\mathbf{W}_B(n)$ for the adaptive filter 110 (FIG. 1) responsive to the outputs of buffers 314 and 318. Adaptive filter 310 is a second adaptive filter and generates an echo estimate supplied to subtractor 312. Subtractor 312 outputs the error signal used by coefficient updating circuit 320.

The coefficient updating circuit 320 continuously adapts, both in a double-talk condition and in a single-talk condition. The coefficient updating circuit 320 provides coefficients from which double-talk detection can be performed. The adaptation of coefficients $\mathbf{W}_C(n)$ is performed as follows:

$$\mathbf{W}_C(n) = \mathbf{W}_P(n) + \mu\, e_1(n)\mathbf{X}(n)[\mathbf{X}(n)^T\, \mathbf{X}(n)]^{-1} \qquad (3)$$

where μ is a step size and $\mathbf{W}_C(n) = [w^C{}_0(n)\ w^C{}_1(n)\ \ldots\ w^C{}_{L-1}(n)]^T$ are the current coefficients. During a single-talk condition, the coefficients $W_B(n)$ are continuously updated to coefficients $\mathbf{W}_C(n)$ for inputting to the adaptive filter 110 through the output 407.

For adaptation, the current coefficients $\mathbf{W}_C(n)$ are stored as previous coefficients $\mathbf{W}_P(n)$ in buffer 314 just prior to updating in coefficient updating circuit 320. Following adaptation, the new updated coefficients are stored as the current coefficients $\mathbf{W}_C(n)$ in buffer 318. The buffers 314 and 318 can be implemented using any suitable memory device, such as parallel load shift

registers, random access memory (RAM), electronically alterable read only memory (EEPROM) or the like.

The control device 300 generates an echo estimate $z_1(n)$ using the previous filter coefficients $W_P(n)$. The echo estimate is generated by adaptive filter 310 using the coefficients $\mathbf{W}_P(n)$ of the previous sampling period and the current far-end signal x(n) to generate the echo estimate $z_1(n)$ as follows:

$$z_1(n) = \mathbf{W}_P(n)^T \mathbf{X}(n) = \sum_{i=0}^{L-1} w^P_i(n)\, x(n-i) \qquad (4)$$

where $\mathbf{W}_P(n) = [w^P_0(n)\ w^P_1(n)\ \ldots\ w^P_{L-1}(n)]^T$ are the previous filter coefficients.

The control device 300 further includes a subtractor 312 to generate an echo estimate error signal $e_1(n)$ for use in coefficient adaptation within control device 300. The echo estimate error signal at output 303 is generated as follows:

$$e_1(n) = y(n) - z_1(n). \qquad (5)$$

The control device 300 further includes a double-talk detection device 400 to perform double-talk detection and output updated filter coefficients $\mathbf{W}_B(n)$ for adaptive filter 110 (shown in FIG. 1). Double-talk detection device 400 (FIG. 2) receives an input $\mathbf{W}_C(n)$ from output 401 and an input $\mathbf{W}_P(n)$ from an output 403, and outputs an updated coefficients $\mathbf{W}_B(n)$ at output 407.

The double-talk detection device 400 (FIG. 3) receives the previous filter coefficients $\mathbf{W}_P(n)$ from output 403 and the current filter coefficients $\mathbf{W}_C(n)$ from output 401, and outputs the filter coefficients for adaptive filter 110 either at output 413 or output 415. During the double-talk condition, the coefficients in buffer 411, and thus the coefficients supplied to the adaptive filter 110, will not change.

The double-talk detection device 400 includes a switch 402 that selects between output 501 and output 503 based upon a state of a double-talk flag DF (shown in FIG. 4 and FIG. 5). DF is a double-talk flag that indicates the current status of the double-talk detection device 400 wherein, DF is 1 in the double-talk condition and DF is 0 in the single-talk condition (a single-talk condition for the purposes of this flag is all times that a double-talk condition does not exist).The previous filter coefficients $\mathbf{W}_P(n)$ and the current filter coefficients $\mathbf{W}_C(n)$ are at output 501 if a double-talk condition is not detected. If a double-talk condition is detected, the output 503 is selected and

outputs both the previous filter coefficients $\mathbf{W}_P(n)$ and the current filter coefficients $\mathbf{W}_C(n)$.

The double-talk detection device 400 (FIG. 3) includes a buffer 411 to store filter coefficients $\mathbf{W}_B(n)$ for the adaptive filter 110 (FIG. 1). The coefficients $\mathbf{W}_B(n)$ in buffer 411 are selectively updated by double-talk detector 600 and single-talk detector 800, and are output through the output 407, continuously regardless of single-talk or double-talk condition, to the adaptive filter 110 (FIG. 1).

The double-talk detection device 400 (FIG. 3) further includes a double-talk detector 600 to perform double-talk detection and update the filter coefficients $\mathbf{W}_B(n)$ in the buffer 411, and a single-talk detector 800 to perform single-talk detection and update the filter coefficients $\mathbf{W}_B(n)$ in the buffer 411. As described herein, the double-talk detector outputs a double-talk flag at output 409 and sets the double-talk flag DF to one when a double-talk condition is detected. The single-talk detector 800 outputs the double-talk flag DF to the switch 402 and resets the double-talk flag DF to 0 when a single-talk condition is detected. The double-talk detector 600 outputs the current coefficients to update the adaptive filter coefficients $\mathbf{W}_B(n)$ in the buffer 411 for the adaptive filter 110 (FIG. 1) in a single-talk condition through the output 413. In the double-talk condition, the single-talk detector 800 updates the adaptive filter coefficients $\mathbf{W}_B(n)$ in the buffer 411 for the adaptive filter 110 (FIG. 1) by copying the current coefficients $W_C(n)$ once a single-talk condition is detected through the output 415. Coefficients $\mathbf{W}_B(n)$ are always the same coefficients in a double-talk condition, whereas the double-talk detector constantly outputs updated coefficients in the single-talk condition.

Referring next to FIG. 4, operation of the double-talk detector 600 will now be described. To perform double-talk detection under single-talk condition, double-talk detector 600 receives from output 501(FIG. 3) both the previous filter coefficients $\mathbf{W}_P(n)$ and the current filter coefficients $\mathbf{W}_C(n)$. The double-talk detector 600 calculates in step 610 the rate of change of the variance of the filter coefficients by calculating a squared norm D(n) of the difference of $\mathbf{W}_C(n)$ and $\mathbf{W}_P(n)$ as follows:

$$D(n) = ||\mathbf{W}_C(n) - \mathbf{W}_P(n)||^2 \qquad (6)$$

where the above squared norm can be expressed as follows:

$$\|\mathbf{W}_C(n) - \mathbf{W}_P(n)\|^2 = \sum_{i=0}^{L-1} [\, w^C_i(n) - w^P_i(n)]^2 \qquad (7)$$

L being the number of coefficients in the adaptive filters 110 (FIG. 1) and 310 (FIG. 2), such that a sum of the differences of the coefficients in each of the filter positions is generated. Although the filters can have the same number of taps, or coefficients, those skilled in the art will recognize that they can have a different number of taps, in which case L will be the number of taps in adaptive filter 310. It will also be recognized by those skilled in the art that the variance can be determined by other means. The present invention reliably detects a double-talk condition by recognizing that the coefficients will oscillate sharply during adaptation in a double-talk condition, whereas the variance of the coefficients of the adaptive filter becomes invariant after convergence in the single-talk condition regardless of the near-end noise. Thus the present double-talk detector can operate effectively even in a noisy environment.

It is determined at step 612 whether the D(n) is greater than a variance threshold ATH(n), wherein ATH(n) is an adaptive threshold. The updating procedure can for example be low-pass filtering in single-talk condition using the following equation for ATH(n):

$$ATH(n) = (1-\alpha)\, ATH(n) + \alpha\, D(n) \qquad (8)$$

where α is a scalar between 0 and 1, and may for example have a value of 0.01. Alternatively, the average value of D(n) can be used as the adaptive threshold. However, the low-pass filtered threshold is desired. An adaptive threshold is used because the variance of the coefficient is almost constant in a single-talk condition after convergence regardless of the presence of near-end noise. Thus the adaptive threshold will converge from an initial value as the adaptive filter converges regardless of background noise. Once convergence occurs, the threshold will be almost a constant. Additionally, the adaptive threshold will permit a higher double-talk threshold if the background noise has a higher average level. This permits the double-talk detector to operate reliably in various noise conditions.

Referring again to FIG. 4, if the squared norm D(n) is greater than the adaptive threshold ATH(n), then the double-talk detector 600 increments the double-talk counter DT_CNT by one in step 616 to indicate the number of times the squared norm D(n) sequentially exceeds the adaptive threshold

ATH(n). If the double-talk counter DT_CNT exceeds a double-talk threshold DTH, as determined in step 618, the double-talk condition is detected. The double-talk flag DF is set to a high logic level and the counter ST_CNT is set to zero in step 620. The double-talk flag DF is outputted to an output 409 and the filter coefficients $\mathbf{W}_B(n)$ are outputted to an output 413 in step 624.

If the double-talk counter DT_CNT is not larger than a double-talk threshold level DTH, as determined at step 618, no double-talk condition is detected. In this case, the double-talk flag DF is set to 0 in step 630. It is envisioned that the double-talk threshold will be short, such as 50 sampling intervals, or 0.01 seconds, such that the double-talk condition is detected quickly. The double-talk flag DF is outputted to an output 409 and the filter coefficients $\mathbf{W}_B(n)$ are outputted to an output 413, in step 624.

Referring once again to FIG. 4 in the case where it is determined that the squared norm D(n) is not greater than ATH(n) in step 612, double-talk detector 600 updates the double-talk counter DT_CNT by setting the counter to 0, updates the adaptive variance threshold ATH(n) according to equation (8), and updates the filter coefficients $\mathbf{W}_B(n)$ by copying either $\mathbf{W}_C(n)$ or $\mathbf{W}_P(n)$, as indicated in step 628. The double-talk detector 600 further in step 630 to set the double-talk flag DF = 0 and output the double-talk flag DF from output 409 and output the coefficients $\mathbf{W}_B(n)$ from output 413.

In summary, the double-talk counter DT_CNT is an integer counter for double-talk detection, DTH is a double-talk threshold level for the double-talk counter DT_CNT. Described below are a single-talk counter ST_CNT, which is an integer counter for single-talk detection, and STH which is a single-talk threshold level for the single-talk counter ST_CNT.

Referring next to FIG. 5, the operation of single-talk detector 800, which performs single-talk detection during a double-talk condition, will now be described. Single-talk detector 800 receives both the previous filter coefficients $\mathbf{W}_P(n)$ and the current filter coefficients $\mathbf{W}_C(n)$ from output 503 . The single-talk detector 800 calculates in step 610 the square of the norm of the difference of $\mathbf{W}_C(n)$ and $\mathbf{W}_P(n)$ as defined with respect to double-talk detector 600.

Referring again to FIG. 5, if it is determined in step 712 that the squared norm D(n) is smaller than the adaptive threshold ATH(n), the single-talk counter ST_CNT is incremented by one in step 806. It is then determined in step 807 whether the single-talk counter ST_CNT is larger than a single-

talk threshold STH. If the single-talk counter ST_CNT is greater than the single-talk threshold STH, a single-talk condition is detected. It is envisioned that single-talk threshold is relatively large, and thus can for example be 2000 sampling intervals, or 0.25 seconds. Upon detecting a single-talk condition, the double-talk flag DF is set to 0, and the double-talk counter DT_CNT is set to zero in step 810. The double-talk flag DF is output to an output 405 and the filter coefficients $\mathbf{W}_B(n)$ is updated by copying the current coefficients $\mathbf{W}_C(n)$ which are outputted through output 415 in step 811 such that these current coefficients are stored in buffer 411.

If it is determined in step 807 that the single-talk counter ST_CNT is not greater than the single-talk threshold STH, no single-talk condition is detected. In this case, the double-talk flag DF is set to 1 in step 812. The double-talk flag DF is outputted to an output 405, and the filter coefficients $\mathbf{W}_B(n)$ are not output to an output 415.

Referring once again to FIG. 5, if the squared norm D(n) is greater than ATH(n) as determined in step 712, then the single-talk counter ST_CNT is set equal to 0 in step 804 and the double-talk flag DF is set to 1 in step 812. The double-talk flag DF is generated at output 405 and the filter coefficients $\mathbf{W}_B(n)$ are not output through output 415.

It can thus be seen that a new double-talk detector uses the measurements of the variance of the coefficients of the adaptive filter to make a double-talk decision. Additionally, a variable threshold is employed to accommodate various noise conditions at the near-end. Consequently, the double-talk detector is largely insensitive to near-end noise and robust in a noisy environment. Because double-talk detection device measures the variance directly from the adaptive filter, it is precise and reliable.

CLAIMS

1. A method of detecting presence of near-end voice in an echo canceller for cancelling an echo of a far-end signal present in a near-end signal, the method comprising the steps of:

updating filter coefficients in an adaptive filter (310) coupled to receive the far-end signal and an error signal;

calculating (610) a variance of the filter coefficients; and

detecting (618) a double-talk condition if the variance of the filter coefficients exceeds a variance threshold during a single-talk condition.

2. The method as defined in claim 1, further including the step of detecting (618) a single-talk condition when the variance of the filter coefficients drops below the variance threshold during a double-talk condition.

3. The method as defined in claim 2, wherein the single-talk condition is detected when the variance is below the variance threshold for a predetermined time period (616, 618).

4. The method as defined in claims 1 or 2, further including the step of adaptively generating the variance threshold.

5. The method as defined in claims 2 or 4, wherein the step of detecting a single-talk condition includes the step of generating a summation signal of a square of differences of the filter coefficients.

6. The method as defined in claim 5, wherein the variance threshold is generated by averaging the summation signal.

7. The method as defined in claim 5, wherein the variance threshold is generated by low-pass filtering the summation signal.

8. The method as defined in claim 1, wherein the double-talk condition is detected when a variance rate of change of filter coefficients exceeds the variance threshold for a predetermined time period (806, 807).

9. The method as defined in claim 8, wherein the step of detecting a double-talk condition includes summing a square of a difference between a most recent coefficient and a previous coefficient in a coefficient position.

10. The method as defined in claim 9, wherein summing includes:

calculating a difference for each coefficient position of the adaptive filter;

squaring each difference; and

summing the square the differences for all of the coefficient positions.

11. A control device (300) performing the method according to any one of claims 1-10.

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Claims searched:	all	**Date of search:**	9 February 2000

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Databases searched:

UK Patent Office collections, including GB, EP, WO & US patent specifications, in:

UK Cl (Ed.R): H4R (RLES)

Int Cl (Ed.7): H04B (3/23); H04M (9/08)

Other: Online: EPODOC,WPI,JAPIO

Documents considered to be relevant:

Category	Identity of document and relevant passage	Relevant to claims
X	EP 0708535 A (NTT) see page 7, line 15 to page 8, line 44	1,11

X	Document indicating lack of novelty or inventive step	A	Document indicating technological background and/or state of the art.
Y	Document indicating lack of inventive step if combined with one or more other documents of same category.	P	Document published on or after the declared priority date but before the filing date of this invention.
&	Member of the same patent family	E	Patent document published on or after, but with priority date earlier than, the filing date of this application.